REMARKS

A set of formal drawings are being submitted herewith. The specification has been amended to correct grammatical mistakes and to make reference numbers consistent with the figures. No new subject matter has been added. Claims 1-17 remain in the subject application. Claims 1 and 17 have been amended, as recited hereinabove.

Claims 1-13 and 17 have been rejected, claims 14-16 have been allowed. Claims 1, 5-13 and 17 have been rejected under 35 U.S.C. 102(e) as being anticipated by Vargo et al. (U.S. Patent No. 6,356,545).

It is stated on page 2, paragraph 1 of the office action that with respect to claims 1 and 17, the limitations recited in clami 1, a "DSP module responsive to an analog signal from one of the telephone devices … and further operative to packetize digital telephone signal for transmission to a …" are inherently addressed by Vargo. Applicant respectfully disagrees with the same, as there is no teaching of such a DSP module by Vargo. Additionally, the DSP module of the claimed invention is discussed on page 7 , lines 22-25 of the subject application as including "a number of DSP chips (or integrated circuits), which are special purpose processors for efficiently executing mathematical operation, such as multiply and add operations, in one clock cycle." Again, such a DSP module does not appear to be taught by Vargo.

Furthermore, claims 1 and 17, as amended recite "initially negotiating a first type of codec", which is not taught by Vargo.

It is believed that claims 1 and 17, as amended, are not anticipated by Vargo and therefore patentable. It is further believed that all claims depending therefrom are patentable. Reconsideration and allowance of the same is respectfully requested.

Applicants submit that the application is now in condition for allowance and an early notice thereof is requested. Should any further amendment be required prior to passing the application to issue, the Examiner is respectfully invited to contact the undersigned by telephone at the number set out below.

Respectfully submitted,

Maryam Imam
Reg. No. 38,190

Dated: September 11, 2002

LAW OFFICES OF IMAM
111 North Market Street, Suite 1010
San Jose, CA 95113
Tel: 408-271-8752/3

Fax: 408-271-8886

I hereby certify that this correspondence with all attachments is being deposited with the U.S. Postal Service as first class mail in an envelope addressed to: Box No Fee Amendment, Assistant Commissioner for Patents, Washington, D.C. 20231 on September 11, 2002 by Erika Villafana.

Erika Villafana

VERSION WITH MARKINGS TO SHOW CHANGES MADE

In the Specification:

Please amend the specification as follows:

Please replace the paragraph starting on page 7, line 1 and ending at page 7, line 6 with the following paragraph:

--The exchange 304 and router 306 are typically situated locally with respect to the telephone 302 and the exchange 312 and router 310 are situated locally with respect to the telephone device 314. But eh telephone 302, the exchange 304 and the router 306 are located remotely to the router 310, the exchange 312 and the telephone device 314. While not shown in Fig. 3, it sould be understood by those skilled in the art that the exchanges 304 and 312 are typically coupled to many telephone devices.--

Please replace the paragraph starting on page 8, at line 24 and ending at page 9, line 2 with the following paragraph:

--The packet rate is proportional to the size of the packet (the higher the bit-rate, the higher the packet size). The packet size after compression is a function of the sophistication of the particular compression algorithm. As noted above, for example, the G732.1 technique yields a smaller packet size than the G.711 compression technique. While a smaller packet size increases system capacity and throughput, the quality of voice is compromised. Thus, the G.711 is likely to yield [the] a better quality voice transmission than the G723.1. It should be noted that whichever codec is utilized by the router 306 is also used by the router 310 to de-compress the transmitted packets.--

In the Claims:

Please amend the following claims:

1. (Once Amended) A router device for use in a communication system having at least two telephone devices in communications with each other for transferring voice information therebetween through a packet switching network, the router device being coupled between one of the telephone devices and the packet switching network and for

performing one of a plurality of types of compression/decompression (codec) operation on information being transferred between the telephone devices comprising:

a Digital Signal Processor (DSP) module responsive to an analog telephone signal from one of the telephone devices and operative to convert the analog telephone signal to a digital telephone signal and further operative to packetize the digital telephone signal for transmission to a remotely-located router device, the router device and the remotely-located device initially negotiating to utilize a first type of codec, the DSP module for switching from using [a] said first type of codec to using a second type of codec upon detection of degradation in the quality of the voice information,

wherein switching between the codecs is performed while a conversation is taking place between the two telephone devices yet avoiding substantial disturbance to users of the telephone devices.

17. (Once Amended) A method for use in a communication system having at least two telephone devices in communications with each other for transferring voice information therebetween through a packet switching network, the router device being coupled between one of the telephone devices and the packet switching network and for performing one of a plurality of types of compression/decompression (codec) operation on information being transferred between the telephone devices comprising:

receiving an analog telephone signal through a telephone connection from one of the telephone devices;

converting the analog telephone signal to a digital telephone signal;

separating information carried on the digital telephone signal into packets of information;

initially negotiating a first type of codec for communication between the telephone devices;

using [a] the first type of codec for transferring the packets of information between the two telephone devices through the packet switching network; and

switching to using a second type of codec upon detection of degradation in the quality of the voice information during the course of the telephone connection.